B4 cont'd

(3) playing said first one of said acoustic musical instruments to produce sounds as directly picked up by said first and second microphones;

(4) comparing with a processor signals from said first and second microphones; and

(5) creating a first digital filter algorithm with said processor to match the signal from said first microphone with the signal from said second microphone.

B5

19. (Amended) The method of claim 16 further comprising:

(6) repeating steps (1) to (5) for a second one of said acoustic musical instruments to create a second digital filter algorithm; and

(7) averaging the results of the first and second digital filter algorithms and creating a new digital algorithm.

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24. (Twice Amended) The system of claim 22 further comprising:

a third microphone [place] placed proximately to a third type of said acoustical musical instrument; and

a digital signal processor coupled to said third microphone and adapted to average results of [apply] said digital filter algorithms [averaged by said processor] into a new digital algorithm and apply said new digital algorithm to signals from said third microphone.

REMARKS

Claims 1-25 and 27 remain in the application. Claims 1, 6, 13, 16, 19, and 24 have been amended. Claim 6 has been amended, in part, to broaden the timing of recordings for the first

and second microphones.

Claim Rejections Under 35 U.S.C. § 112, second paragraph

Claims 19, 20, 23 and 24 were rejected as failing to distinctly claim the invention under 35 U.S.C. § 112, second paragraph. Specifically, the Office Action takes the position that "algorithms cannot be averaged." Though Applicant respectfully disagrees with that conclusion, claims 19 and 24 have been amended to refer to the results of algorithms being averaged to create a new algorithm. In view of these amendments, reconsideration and withdrawal of the rejections of claims 19, 20, 23, and 24 under 35 U.S.C. § 112, second paragraph is respectfully requested.

Rejections Under 35 U.S.C. §§ 102(e) and 103 (a)

Claims 25 and 27 were rejected under 35 U.S.C. § 102(e) as being anticipated by U.S. Patent No. 5,714,918 to Menkhoff ("Menkhoff"). Claims 1-5 were rejected under 35 U.S.C. § 103(a) as unpatentable over U.S. Patent No. 4,340,780 to Odlen ("Odlen"). Claims 1, 6-11, 13-18, 21-22 were rejected under 35 U.S.C. § 103(a) as unpatentable over U.S. Patent No. 5,506,910 to Miller et al. ("Miller"). Claim 12 was rejected under 35 U.S.C. § 103(a) as being unpatentable over Miller in view of U.S. Patent No. 5,537,614 to Hagimori ("Hagimori"). These rejections are traversed, in part, because the cited references fail to teach or suggest a microphone-tailored equalizing system and method as recited in the pending claims.

Claims 25 and 27 and Menkhoff

In the Office Action of February 1, 2000, claims 25 and 27 were rejected under 35 U.S.C. § 102(e) as being anticipated by Menkhoff. Applicant has reviewed this reference and maintains

that Menkhoff fails to teach or suggest that a frequency pole of a high-pass filter and a frequency pole of a low-pass filter are controlled independently of each other as recited in these claims (an argument presented in his Amendment of June 1, 2000). Though the Examiner maintains that he is unpersuaded by this argument, the current Office Action is silent as to where this feature is shown in Menkhoff. If this rejection is to be maintained, Applicant respectfully requests that the Examiner point out with some level of specificity where this feature is shown in Menkhoff. In Fig. 6 of Menkhoff, the control unit st controls the amount of the <u>output</u> signals from the high and low pass filters are passed to adder ad1. There is no disclosure in Menkhoff (especially Fig. 6) for controlling the <u>poles</u> of the high and low pass filters as recited in the claims. Reconsideration and withdrawal of the rejection of claims 25 and 27 under 35 U.S.C. § 102(e) is respectively requested.

<u>Claims 1-5 and Odlen</u>

Claim 1 refers to a comparison between sounds from an acoustic instrument (e.g., a guitar). The comparison is made between sounds of the acoustic instrument as picked up by a first microphone and reference sounds of the acoustic instrument. "Reference sounds" is a term defined in the specification (see Pages 8-9). Claim 1 (as well as claims 6, 13 and 16) has been amended to refer to reference sounds as coming directly from the acoustic instrument. Once these sounds are compared, a tailor-made equalizer can be created to compensate for difference between the two. In one embodiment, if the reference sounds are detected at an "optimum" location, the output of the first microphone will be similar to the reference sounds (i.e., will sound as if it is picking up the acoustical musical instrument at an optimum location). In claim 5, this feature of the present invention is taken a step further in that several acoustic instruments

of a given type are used so as to create a tailor-made equalizer that would work with the first microphone on any other acoustical instruments of the same type.

Odlen does not show or suggest these features of the invention. The teaching of Odlen is to compare the output of a speaker, via a listening environment, with the output of a "source." There is no teaching in Odlen that would suggest the presently claimed invention - the comparison of sounds at two different positions of the same acoustical instrument and the creation of a tailor-maid equalizer to compensate for that difference. First, Odlen does not even mention an acoustic instrument, only a "source" 26. This source, necessarily, outputs an electric signal for input to the averaging circuit 40 and the attenuator 30. Claim 1 specifically recites the comparison of sounds picked up with a first microphone and reference sounds <u>directly from</u> the acoustic instrument. In Odlen, no reference sounds (as defined in the specification and recited in the claims) are being compared to the output of source 26. Instead, the output of source 26 is fed through acoustic generator 28 (i.e., a speaker) to be picked up by the acoustic detector 38 (i.e., a microphone). There is nothing in Odlen that teaches or suggests using the acoustic detector 38 to receive reference sounds of an acoustic instrument (even assuming that source 26 includes an acoustic instrument). Odlen is specifically directed to correcting the effects of a listening environment on the output of a speaker (see Col. 2, lines 31-34). The present invention in claims 1-5 concerns correcting the output of a first microphone placed proximately to an acoustic instrument based on differences between those sounds and sounds directly from the acoustic instrument and not a speaker.

Moreover, the system of Odlen will not work with an acoustic source. For the setup in Odlen, the speaker 28 is relatively close to the microphone at Source 26 and not the microphone at 38. The speaker will emit sound at a much louder volume than the acoustic signal from the

source itself (which is the purpose of a speaker system to begin with). Thus, the microphone at Source 26 will be greatly contaminated by the output of the speaker 28. Because of this contamination, it will be impossible for the system of Odlen to compare the acoustic sound from an instrument at two different locations.

Claims 1, 6-18, and 21-22 and Miller

In addition to claim 1, independent claims 6 and 13 refer to a comparison of sounds from a first microphone placed proximate to an acoustic musical instrument and reference sounds directly from the instrument. Claim 21 refers to first and second microphones placed at selected distances from an acoustic musical instrument. In claim 6, a tailor-made equalizer is designed based on a comparison of the sounds. In claims 13 and 21, a digital filter algorithm is created for application by a processor to compensate for differences between the sounds.

Miller fails to teach or suggest these features of the independent claims. In Miller, the input signal (e.g., a musical instrument 26, a microphone 28 or a recorded program 30) is passed into a narrow band reject filter 21 (sometimes referred to as a "notch" filter). In such a filter, a frequency band is removed from the input signal (see Col. 4, lines 47-53). A sine-wave generator 22 is used to generate a new signal to take the place of the sound removed from that input into the notch filter. The resulting signal is output by a speaker 36 and picked up by a microphone 40. A sine wave detector 42 detects the signal from the microphone that was originally generated by the masked sine wave adder 22. Based on a comparison between the measured amplitude level and a desired level, a compensation can be made in the multi-band gain control 32.

In the Office Action it states at page 6 that "[a]s disclosed in column 3, lines 32-60, the

reference microphone 40 picks up sounds made by an instrument and the sine wave adder and adjusts the multi-band gain control 32 to produce a desired frequency response." The Applicant finds this to be an incomplete characterization of what is described in Miller. Any music that is generated at mixer/preamplifier 24 is NOT compared to anything, and is not used to adjust the multi-band gain control 32. As explained above, a frequency band of the output of mixer/preamplifier 24 is removed from the signal output by mixer/preamplifier and replaced by the signal generated by the masked sine wave adder. The music and the added sine wave are output by speaker 36, but only the added sine wave is output by sine wave detector 42. Note that the signal "FREQUENCY SELECT" is output by control 44 and input to notch filter 21, sine wave adder 22, and sine wave detector 42 in Fig. 1. Thus, any comparison that is performed in Miller is between the following:

1. The sine wave added by element 22, output by speaker 36 and received at sine wave detector 42; and
2. a "desired response," which appears to be a preset constant in Fig. 2.

The pending claims require a comparison between sounds from an acoustic musical instrument received at a first microphone and sounds from the musical instrument received at a second microphone. As can be seen from Miller, such a comparison is not contemplated. More importantly, such a comparison is neither shown, taught, nor suggested by Miller. In fact Miller specifically teaches away from the presently claimed invention, in that Miller <u>dispenses</u> with the sound from the musical instrument and adds a controlled sine wave for comparison purposes.

The Examiner also states on page 7 that "one would have been motivated to use their system to generate a response which is the natural sound of the instrument" rather than a flat response. That is, any desired response could be preprogrammed in the system before the device

acts on the signal. However, the present invention teaches a process for determining just such a "response which is the natural sound of the instrument." Since Miller can only compensate for differences between the output of the Automatic Equalizer 20 and the microphone at 40, not for any differences between the input to the Automatic Equalizer 20 and the microphone at 40, the argument that the desired response could be a response which is the natural sound of the instrument is a circular argument.

Summary

Though it is true that Odlen and Miller disclose collecting two signals, comparing them, and making a change in attenuation, the references disclose systems and methods that are different from those presently claimed. In order to justify a rejection of the claims under 35 U.S.C. § 103(a), there must be some suggestion in the references or prior art to modify them to achieve the presently claimed invention. (See MPEP § 2143). The differences between the references and the claimed inventions are not trivial, in part because they are drawn to completely different issues.

In the prior art, the room where music will be performed will have an effect on the sound emanating from the stage, for example. To compensate for any negative effects, the outputs from the microphones would be passed through a sound board and modified depending on the effects of the room on the music output by the speakers. To do this, testing sounds would be generated (e.g., the musicians would play, or recorded material would be played, or "pink" noise would be generated). The sound board would be adjusted accordingly to compensate for any negative effects of the room. Unfortunately, all of that work could be wasted because the addition of an audience to the room has a dramatic effect on sound quality. Odlen compensates for this by

comparing the output of the speakers with the output of a microphone, and making adjustments to the speaker output. Miller compensates for this by notching out a frequency band of sound to be played through the speakers, replacing the sound with a generated reference signal and comparing the detected reference signal with a desired bench mark so that modifications can be made to signals before being output by the speakers. Most importantly, the circuits and methods of Odlen and Miller may be used <u>during</u> a performance, while the audience is present, thus eliminating the guesswork that would be necessary during the sound checks discussed above.

The present invention concerns a different issue - how to make a microphone output a signal that sounds like it is being placed at a more desirable position. In the case of a guitar, the reference position may be six feet in front of the sound hole of the guitar. Placing a microphone in such a location is impossible in a performance situation, because the sound level of the guitar reaching a microphone at that distance would be overwhelmed by the sound level coming out of the speaker system, resulting in a contaminated source signal, and often resulting in feedback (called 'feedback-to-gain ratio'). The present invention compensates for this by placing the microphone proximate to the guitar (for example on the guitar) and comparing, preferably in a controlled test environment, the sounds from the microphone with reference sounds received directly from the guitar (e.g., with a microphone placed at an optimal location). Once the comparison is done, and the effects on the sound of the instrument caused by placing the microphone proximately to it are determined, a tailor-made equalizer for that microphone and instrument can be made that will make the sound from that microphone sound as if it was coming from a more optimal location. If several instruments of the same type are measured in the above manner, the effects can be averaged to give a standard equalizer for all instruments of that type. The effects of the present invention are not taught or suggested by the cited references.

Accordingly, reconsideration and withdrawal of the rejection of claims 1-18 and 21-22 under 35 U.S.C. § 103(a) is respectfully requested.

CONCLUSION

For all the above reasons, the Applicant respectfully submits that this application is in condition for allowance. A Notice of Allowance is earnestly solicited.

The Examiner is invited to contact the undersigned at (408) 975-7500 to discuss any matter concerning this application. The Office is hereby authorized to charge any additional fees or credit any overpayments under 37 C.F.R. § 1.16 or § 1.17 to Deposit Account No. 11-0600.

Respectfully submitted,
KENYON & KENYON

Dated: 2/28/01

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